An Open-Loop Class-D Audio Amplifier with Increased Low-Distortion Output Power and PVT-Insensitive EMI Reduction

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Abstract — An open-loop class-D audio amplifier with the proposed adaptive-coefficient delta-sigma modulator (ACDSM) and low-EMI control method is implemented in a 0.18 μm BCD process. Compared with conventional DSMs, the ACDSM simultaneously achieves a wide stable input range and high in-band noise suppression, resulting in a 20% increase of low-distortion output power for open-loop class-D amplifiers without boosting the supply voltage or degrading the THD+N at small output power. Moreover, the proposed low-EMI control method realized by a simple finite state machine eases the PVT-sensitive issues of common-mode EMI reduction.

Index Terms — Delta-sigma modulation, electromagnetic interference, power amplifiers, pulse width modulation, sigma-delta modulation, stability.

I. INTRODUCTION

Digital signal processing and digital audio interfaces are widely applied in consumer electronics, driving the demand for class-D audio amplifiers with digital input capability. Although several closed-loop digital-input architectures [1][2] suppress the power stage distortion and supply noise with additional analog blocks, the open-loop digital-input architectures commonly adopted in TVs and home theaters are mostly digitally implemented, allowing for lower complexity, smaller chip area and easier design porting to more advanced processes [3][4].

Figs. 1 (a) and (b) show typical closed- and open-loop architectures, while (c) and (d) plot their corresponding performance at DSMOUT, PWMOUT and power output, respectively. High-order delta-sigma modulators (DSMs) can be used to reduce the required clock rate [3]; however, assuming k=1 in Figs. 1 (a) and (b), if DSM input magnitude exceeds its maximum stable input, DSM instability may lead to degraded SNDR at DSMOUT and PWMOUT; thus the total harmonic distortion plus noise (THD+N) at large output power rapidly increases, as shown by the black solid lines on the right-most plots of Figs. 1 (c) and (d). Consider the typical closed-loop architecture in Fig. 1(a). DSM instability can be prevented by scaling down the interpolator’s gain by k to reduce the DSM input magnitude; then, by scaling the feedback factor to increase the closed-loop gain of the feedback loop, the low-distortion output power, defined as the maximum output power with THD+N<1%, can be increased, as shown by the dashed red lines on the right-most plot of Fig. 1 (c). However, for the typical open-loop architecture in Fig. 1(b), the gain-scaling method to prevent low-distortion output power from degradation by DSM can not be used, since the PWM duty at PWMOUT is clipped at 100% by the high-frequency DSM-shaped noise if the gain of PCM-to-PWM converter is scaled up by 1/k. Therefore, for typical open-loop architectures with given supply voltages and nominal impedances of typical speakers, the low-distortion output power is limited by the DSM maximum stable input instead of the power stage distortion and supply noise, both of which can be reduced.
class-D amplifiers; however, the realization of CMFBBD modulation in [7] is sensitive to both process-voltage-temperature (PVT) variation and varying speaker impedance, resulting in shoot-through currents or degraded EMI reduction. Thus, a PVT-insensitive low-EMI control method is proposed in this work.

II. PROPOSED TECHNIQUES AND IMPLEMENTATION

Fig. 2 shows the block diagram of this work. A digital input signal is fed into the interpolator, and processed by the proposed ACDSM, whose coefficient set \( \text{Set}_n \), comprising \( g_1[n], ..., g_5[n], a_1[n], ..., a_5[n], b_1[n], ..., b_4[n] \), is adaptively changed according to ACDSM input \( x[n] \). Then, the PCM-to-PWM converter generates the PWM signal, which is amplified by the power stage to drive the speaker load after low-pass filtering. The output stage is controlled by the SEL signal to operate in BD-modulation or low-EMI modes, where the former is for high-performance application, and the latter is for cost-effective application since in using CMFBBD modulation, the low-pass L-C filter can be replaced by a bead-C filter or even eliminated.

A. Adaptive-Coefficient Delta-Sigma Modulator (ACDSM)

Fig. 3 shows the noise transfer functions and root-loci plots of two different DSM designs. Coefficient set A \( (\text{Set}_A) \) is designed for a conventional DSM\(_A\) with high in-band noise suppression but smaller maximum stable input; in contrast, coefficient set B \( (\text{Set}_B) \) is designed for a DSM\(_B\) by the RLUIUC method [6] with full-scale stable input but degraded in-band noise suppression. The proposed ACDSM shown in Fig. 4 achieves the advantages of both DSM\(_A\) and DSM\(_B\) by adaptively changing the operating \( \text{Set}_n \) toward \( \text{Set}_A \) or \( \text{Set}_B \) if the coming DSM input is respectively smaller or larger than an input threshold level \( V_{th} \). To avoid severely degraded SNDR due to large coefficient changes, N coefficient sets are linearly interpolated between \( \text{Set}_A \) and \( \text{Set}_B \), allowing \( \text{Set}_n \) to be changed with a smaller step \( \Delta \text{Set}_n = (\text{Set}_A - \text{Set}_B)/(N+1) \) in

![Fig. 2](image-url)  
Fig. 2 (a) Block diagram of this work, and two selectable (b) BD-modulation and (c) low-EMI modes for different applications.

![Fig. 3](image-url)  
Fig. 3 (a) Noise transfer functions and (b) root-loci plots of two different DSM designs.

![Fig. 4](image-url)  
Fig. 4 Proposed ACDSM algorithm and its implementation.
each $x[n]$ sample period. In this design, $V_{ih} = -0.7$dBFS and N=25. The implementation cost for the ACDSM algorithm is insignificant since $\text{Set}[n]$ can be calculated by adding or subtracting the constant $\text{Set}_0$ to or from $\text{Set}[n-1]$. To maintain the same signal transfer function when $\text{Set}[n]$ is adaptively changed, a feed-forward DSM structure with a direct path from $x[n]$ to the quantizer input is used. Fig. 5 shows the SNDR vs. input magnitude plots at the PCM-to-PWM converter output of DSM$_A$, DSM$_B$, and ACDSM, and indicates that both wide stable input range and high in-band noise suppression are achieved by ACDSM.

B. PVT-insensitive low-EMI control method

Fig. 6 shows the presented power stage, and the previous and proposed control methods. In state $S_0$ of the previous method, only $M_{1,4}$ turn on, so $V_{\text{diff}}$ is $+V_{DD}$. Then, when entering state $S_1$, $M_{5,6}$ should turn on at exactly the same time as when $M_{1,4}$ turn off, so that both $\text{OUT}_P$ and $\text{OUT}_N$ are smoothly driven to $V_{DD}/2$ [7]. However, PVT variation of the gate drivers may cause timing skew between $M_{5,6}$ turning-on and $M_{1,4}$ turning-off, resulting in significant shoot-through current and/or additional output voltage transition. By contrast, the proposed method turns on $M_{1,4}$ and $M_5$ when $V_{\text{diff}}$ is driven to $+V_{DD}$ in state $S_A$. Thus, when entering state $S_B$, as soon as $M_{1,4}$ turn off, $I_{\text{Load}}$ flows through a free-wheeling path formed by the speaker load, the body diode of $M_6$, and the turned-on $M_5$. Neither $M_5$ nor $M_6$ need to be turned-on at exactly the same time as when $M_{1,4}$ are turned off; therefore, a smooth transition is achieved in the proposed method despite typical PVT variations. The power stage then enters state $S_C$ by turning on $M_6$ to complete this transition. A careful power stage layout can reduce the substrate-current effect introduced by the free-wheeling path.

III. MEASUREMENT RESULTS

Fig. 7 shows the measured THD+N vs. output power in BD-modulation mode. For a nominal load impedance $R_L$ of 8Ω and a supply voltage $V_{DD}$ of 24V, the measured low-distortion output power $P_{\text{OUT}}$, defined as the maximum output power at THD+N < 1%, of the class-D amplifier with the ACDSM reaches 30W, an increase of 20% compared to that with the conventional DSM$_A$. Fig. 8 shows EMI measurements of the two modes with bead-C output filters on the same flat panel TV. The peak conducted and radiated EMI are respectively reduced by 8 dBμV/m and 24 dBμV/m in the low-EMI mode compared to those in BD-modulation mode. To further demonstrate the increased low-distortion output power contributed by ACDSM under different conditions of $V_{DD}$, $R_L$ and the parasitics of MOSFET $R_{\text{dsON}}$ and metal lines, a normalized output power is defined as $P_{\text{OUT}}[\eta V_{DD}/2/(2 R_L)]$, where $\eta$ is the peak efficiency. Due to the extended +0.2dBFS maximum stable input of the ACDSM, the normalized output power of this work increased by 5W.
reaches +1.03, meaning that the ACDSM can be adopted to increase the low-distortion output power by at least 12% for other state-of-the-art digital-input open-loop class-D amplifiers, as shown in TABLE I. The stereo amplifier of this work is implemented in TSMC 0.18μm BCD process, and the integrated DSM is designed by a conventional method similar to DSMa, while the THD+N in Fig. 7 is measured via the integrated dual-mode power stages inputted by the external PWM codes generated by DSMa, DSMb and ACDSM. The chip micrograph in Fig. 9 shows this work occupies 3.74 mm², and is integrated with a digital audio processor, containing volume control, bass, treble, equalizer, etc.

IV. CONCLUSION

A 30-W open-loop class-D audio amplifier is implemented in a 0.18μm BCD process. With the proposed adaptive-coefficient delta-sigma modulator (ACDSM), the output power with THD+N < 1% is increased by 20% without boosting the supply voltage or degrading the THD+N at small output power. In addition, the PVT-sensitive issues of common-mode EMI reduction are eased by the proposed low-complexity control method.

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REFERENCES